REMARKS

Claims 11-33 are currently pending in the subject application, and are presently under consideration. Claims 11-33 are rejected. Claims 11 and 16 have been amended. Favorable reconsideration of the application is requested in view of the amendments and comments herein.

I. Rejection of Claims 11-13 and 16-18 Under 35 U.S.C. §103(a)

Claims 11-13 and 16-18 stand(s) rejected under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,825,898 to Marash ("Marash") in view of U.S. Patent No. 4,581,758 to Coker et al. ("Coker"). Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claim 11 has been amended and recites a microphone array processing system for performance enhancement in noisy environments comprising a signal summation circuit for combining the adaptively filtered output signals from the microphones, such that signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently to produce an increased signal-to-noise ratio. The amendments to claim 11, being made for the sake of clarity and consistency, does not raise any new issues that would require further consideration and/or searching (see, e.g., claim 16, which already recites that the adaptively filtered signals are combined). At the very least, since the changes are consistent with previously presented claims, Representative for Applicant requests that the amendments be entered for the purposes of appeal.

The Examiner asserts that the main channel matrix and reference channel matrix of Marash (Marash, FIG. 1, reference numbers 3 and 4) read on the signal summation circuit of claim 11 (Office Action dated February 28, 2005, page 3). Neither the main channel matrix nor the reference channel matrix, individually or collectively, correspond to the signal summation circuit recited in claim 11 because Marash fails to teach or suggest that either of the matrices operate to combine adaptively filtered signals, as recited in amended claim 11. While the Office Action dated February 28, 2005, maintains the rejection of claim 11, however, the Examiner responds by now contending, in the Response to Arguments section, that the difference unit of

Marash (FIG. 11C, element 260) reads on the signal summation circuit of claim 11 (Office Action dated February 28, 2005, page 8). As the Examiner does not address the arguments against the previous assertion, but instead presents a new assertion as to the signal summation circuit of claim 11, it is assumed by Representative for Applicant that the argument against the assertion that the main channel matrix of Marash reads on the signal summation circuit of claim 11 was found persuasive. Accordingly, it is respectfully requested that this grounds for rejection be withdrawn.

With regard to the assertion that the summation circuit of claim 11 is taught by the summation implementation of Marash (Marash, FIG. 11C, element 260), as proffered by the Examiner (Office Action dated February 28, 2005, page 8), Representative for Applicant maintains respectful disagreement. Representative for Applicant, in the response filed on June 1, 2004, to the Office Action dated March 1, 2004, argued specific differences between the difference unit (260) taught by Marash and the signal summation circuit recited in claim 11. The Examiner responds to this argument in the Office Action dated February 28, 2005, by stating that:

> [S]ince Marash provides for filtering a signal coming in all directions to produce a signal coming in a specific direction such that the data output from the main channel and reference channels are combined. The capability of the system to subtract the noise components from the speech components provide support for the speech signals combining coherently and noise components combining incoherently. (Office Action dated February 28, 2005, page 9).

This argument, as stated in the Office Action dated February 28, 2005, is respectfully traversed because it fails to provide facts or technical reasoning sufficient to support the obviousness conclusion. Marash does not teach or suggest a summation circuit that coherently combines signal components resulting from the speech source, as recited in claim 11. Instead, Marash teaches that the difference unit (260) subtracts canceling signals from the delayed main channel to generate a digital output signal (col. 5, ll. 10-12). The canceling signals are generated by the adaptive filtering of the reference channels (col. 5, ll. 1-7), which represent signals received from directions other than that of the signal source. Thus, the reference channels of

Marash represent interference signals (i.e. noise components), and hence do not contain voice components (col. 4, ll. 63-67). Because there are no voice components in any of the signals output from the adaptive filters in Marash, the difference unit (260) does perform any coherent combining of signal components resulting from a speech source AND incoherent combining of signal components from noise, as recited in claim 11.

Marash, in addition to not teaching or suggesting a coherently combining signal components resulting from the speech source, further does not teach or suggest the incoherent combination of the signal components resulting from a noise source at a signal summation circuit, as recited in claim 11. The difference unit (260) of Marash does not combine signal components resulting from noise, but instead subtracts the canceling signals generated from the reference channel matrix from the main channel signal. Thus, Marash teaches that the output of the difference unit of Marash does not include signal components resulting from noise because the interference signal components of the main channel signal have been subtracted out from the main signal. Accordingly, as mentioned above, the difference unit does not combine adaptively filtered signals in the manner recited in claim 11. The system of canceling noise components, as taught by Marash, more closely resembles prior art approaches described in the Background of the Invention section of the Present Application, which states, "There are also a number of prior art systems that effect active noise cancellation in the acoustic field. Basically, this technique cancels acoustic noise signals by generating an opposite signal...to cancel the unwanted noise signal." (Present Application, page 2, ll. 8-14). Marash, therefore, does not teach or suggest incoherently combining signal components resulting from a noise source at a signal summation circuit, as recited in claim 11.

Moreover, claim 11 also recites a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone. It is this alignment of data microphone output signals with reference microphone output signals that allows the signal components resulting from the speech source to combine coherently (Present Application, page 11, line 29 through page 12, line 2). The adaptive filters of Marash generate canceling signals that are subtracted from the main signal to

generate an output signal substantially free from an interference signal component because the canceling signals closely track the interference signal components (col. 8, ll. 47-54). Furthermore, the main channel signal of Marash is not an output from a reference microphone, as recited in claim 11. Instead, the main channel signal of Marash is a signal created from the fractional weighted sum of all output signals from the microphones, so long as the sum of such signals is one (col. 6, ll. 1-55). Accordingly, because Marash teaches that the interference signals are subtracted from the main channel signal (the fractional weighted sum of all output signals) to reduce the interference signal component, the adaptive filters are aligning the interference (*i.e.*, noise) components of the reference channel signals with the main channel signal. With such an alignment occurring in the adaptive filters of Marash, the canceling signals are then subtracted from the main channel signal to remove noise from the main channel signal. Therefore, signal-to-noise ratio is increased in the system of Marash based on the subtraction of the noise components to make the output signal free from noise. This differs from claim 11 in that signal-to-noise ratio is increased in the system of claim 11 by coherently combining the voice components and incoherently combining the noise components. It is respectfully submitted that the failure of the Office Action dated February 28, 2005, to address the above argument as set forth in the response filed on June 1, 2004, amounts to an admission by the Examiner that the prior art fails to teach or suggest the plurality of adaptive filters that function in the manner recited in claim 11. Therefore, Marash fails to teach or suggest coherently combining the voice components and incoherently combining the noise components, and thus further does not teach or suggest the signal summation circuit of claim 11.

Additionally, the position proffered in the Office Action fails to appreciate the meanings of the phrases "combining coherently" and "combining incoherently," which are succinctly provided in the Detailed Description of the Preferred Embodiments as follows:

> A source of speech...is assumed to be a point source that does not move, at least not rapidly. Since the noise comes from many directions, it is largely independent, or uncorrelated, at each microphone. The system of the invention sums signals from N microphones and, in so doing, achieves a power gain of N^2 for the signal of interest, because the amplitudes of the individual signals from the microphones sum coherently, and power is proportional to the square of the

> amplitude. Because the noise components obtained from the microphones are incoherent, summing them together results in an incoherent power gain proportional to N. Therefore, there is a signal-to-noise ratio improvement by a factor of N^2/N, or N. (Present Application, page 6, ll. 18-27).

The Present Application further discusses the coherent and incoherent combination in the context of the summation circuit, as recited in claim 11:

> In the summation circuit 18 [FIG. 5], the speech signal contributions from the data microphones are added coherently, as previously discussed, to produce a speech signal proportional to $NS*h_1$, and this signal can be conveniently convolved with the transfer function h_1 to produce a larger speech signal NS. The speech signals, being coherent, combine in amplitude, and since the power of a sinusoidal signal is proportional to the square of its amplitude, the speech signal power from N sensors will be N^2 times the power from a single sensor. In contrast, the noise components sensed by each microphone come from many different directions, and combine incoherently in the summation circuit 18. The noise components may be represented by the summation: $n_1 + n_2 + \ldots + n_N$. Because these contributions are incoherent, their powers combine as N but their root mean square (RMS) amplitudes combine as SQRT N. The cumulative noise power from the N sensors is, therefore, increased by a factor N, and the signal-to-noise ratio (the ratio of signal power to noise power) is increased by a factor N^2/N, or N....Theoretically, if the number of sensors is doubled the signal-to-noise ratio should also double, i.e. show an improvement of 3 dB (decibels). (Present Application, page 10, line 24 through page 11, line 12).

As evident from the above descriptions from the Present Application, the differences between the teachings of Marash and claim 11 and are very distinct. By combining coherently, the signal components resulting from the speech source have their amplitudes combined to result in an increased output power because the power of a sinusoidal signal is proportional to the square of its amplitude. The only combination of signal components resulting from the speech source that is taught by Marash occurs in the creation of the main channel signal in the main channel matrix. As described above, the combination of signal components resulting from the speech source as taught by the main channel signal of Marash is not performed by a signal summation circuit that combines adaptively filtered signals, as recited in claim 11. Instead, the main channel signal of Marash is a signal created from the fractional weighted sum of all output signals from the microphones, so long as the sum of such signals is one (col. 6, ll. 1-55).

Therefore, the signal components resulting from the speech source that form the main channel signal of Marash are not coherently combined, as set forth in claim 11.

The addition of Coker does not cure the above-noted deficiencies of Marash with respect to claim 11. Coker is relied upon in the Office Action to show a teaching of a plurality of bandpass filters for eliminating from the microphone output signals a known spectral band containing noise, as recited in claim 11. Marash and Coker, however, taken alone or in combination, fail to teach or suggest the recitations of claim 11.

For at least the reasons stated above, withdrawal of the rejection of claim 11, as well as claims 12 and 13 which depend therefrom, is respectfully requested.

Claim 16 has been amended for clarity and recites a method of improving detection of speech signals, the method comprising adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and combining the adaptively filtered output signals from the microphones in a signal summation circuit, such that signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio. Accordingly, for substantially the same reasons described above with regard to claim 11, claim 16 is patentable over Marash, alone or in combination with Coker. Withdrawal of the rejection of claim 16, as well as claims 17 and 18 which depend therefrom, is respectfully requested.

For the reasons described above, claims 11-13 and 16-18 are patentable over the cited art. Accordingly, withdrawal of this rejection is respectfully requested.

II. Rejection of Claims 14-15 and 19-20 Under 35 U.S.C. §103(a)

Claims 14-15 and 19-20 stands rejected under 35 U.S.C. §103(a) as being unpatentable over Marash in view of Coker, and further in view of Digital Signal Processing Handbook (1998) ("DSP Handbook"). Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claims 14 and 15 depend from claim 11, which, as described above, should be allowed over the cited art. Accordingly, reconsideration and allowance of claims 14 and 15 are respectfully requested.

Claims 19 and 20 depend from claim 16, which, as described above, should be allowed over the cited art. Therefore, reconsideration and allowance of claims 19 and 20 are respectfully requested.

For the reasons described above, claims 14-15 and 19-20 are patentable over the cited art. Accordingly, withdrawal of this rejection is respectfully requested.

III. Rejection of Claim(s) 21-33 Under 35 U.S.C. §103(a)

Claims 21-33 stand rejected under 35 U.S.C. §103(a) as being unpatentable over Marash in view of the DSP Handbook. Withdrawal of this rejection is respectfully requested for at least the following reasons.

Claim 21 recites a method for improving detection of speech signals comprising converting reference microphone data from a reference microphone to a frequency domain and updating filter weight values with the reference microphone data in the frequency domain. Marash teaches that the outputs of each of the microphones are formed into both a main channel signal and a group of reference channel signals (Marash, FIG. 1). Marash further teaches adaptive filters to generate canceling signals that are subtracted from the main signal to generate an output signal substantially free from an interference signal component because the canceling signals closely track the interference signal components (col. 8, ll. 47-54). The output signal is also used to adjust the adaptive filter weights to further reduce the interference signal component (col. 8, ll. 55-57). Thus, Marash specifically teaches that it is the output signal that updates the adaptive filter weights, the output signal being the difference between the main channel signal, which is a composite of all microphone signals, and the canceling signals, which are the interference signal components. Therefore, Marash does not teach or suggest updating filter weight values with the reference microphone data in the frequency domain, as recited in claim 21.

The addition of the DSP Handbook does not cure the deficiencies of Marash in failing to teach what is set forth in claim 21. The DSP Handbook teaches a general adaptive filtering algorithm for increasing convergence speed. Referring to page 22-17, figure 22.11 of the DSP Handbook, the bottom right of the figure shows the desired response signal d_k (analogous to the main channel signal of Marash) input to an adder, which adds a positive value of the desired response signal d_k to a negative value of the filter output vector y_k to create an error signal e_k. It is this error signal e_k that is converted to the frequency domain using a FFT operation (bottom right of figure 22.11) and that is input to the adaptive filter to update the filter with current weight values. Neither the DSP Handbook nor Marash, taken individually or in combination, teaches or suggests converting reference microphone data to the frequency domain, as recited in claim 21. In particular, the desired response signal (presumably main channel signal of Marash, which is provided to the adaptive filter) is not converted to the frequency domain in the DSP Handbook. Instead, the DSP Handbook teaches that it is the difference of the desired response signal and the filter output vector that is converted to the frequency domain. Thus, any adaptive filtering performed based on the combined teachings of Marash and the DSP Handbook, as suggested in the Office Action, would be performed based on the output signal of Marash, such that no updating of filter weight values can be made with reference microphone data, as recited in claim 21.

Because Marash does not teach or suggest the use of a reference microphone data in its method of adaptive filtering, and because the DSP handbook further fails to teach or suggest converting a reference microphone data to a frequency domain as well as updating the filter weight value with the reference microphone data, the Office Action dated February 28, 2005, has thus failed to set forth facts or technical reasoning to support the conclusion that the combination of the references renders claim 21 obvious. If the Examiner is relying on knowledge of one of ordinary skill in the art, as suggested by the Office Action, Representative for Applicant respectfully requests a detailed explanation regarding the specific understanding or technical principle within the knowledge of one of ordinary skill in the art that would suggest the

combination. The rejection of claim 21, as well as claims 22-27 which depend therefrom, is respectfully requested to be withdrawn.

Additionally, claim 24 recites that the updating filter weight value comprises $W(k+1) = W(k)+\mu(Ref(k) – X(k))*conj(Y)$ where k is the data block number and μ is a small adaptive step constant. The DSP Handbook teaches an equation for calculating the updated filter weight in the frequency domain (page 22-17, equation 22.33) of $W_{k+1} = W_k+\mu X_k^H * E_k$, wherein $E_k = FFT(d_k - y_k)$. In this analysis, it is assumed that the variable X (upper and lower case) in the DSP Handbook corresponds to the variable Y (upper and lower case) in claim 24, and vice verse. It is further assumed that, as appearing on page 22-15 (last paragraph) of the DSP Handbook, capital letters are used to denote the frequency-domain variables and lowercase letters to denote the time-based variables. Rewriting equation 22.33, wherein Ref(k) of claim 24 is assumed to correspond to the frequency domain conversion of d_k, and conj(Y) is assumed to correspond to X_k^H, and substituting E_k as well as X and Y, $W_{k+1} = W_k+\mu(FFT(d_k - y_k)) * conj(X)$. This equation is different from that shown in claim 24, the difference being inherent to the difference in adaptive filtering technique, as described above with regard to claim 21. Because the Fourier Transform is a complex form of the Fourier integral, the term $FFT(d_k - y_k)$ of rewritten equation 22.33 is not equal to the term $(Ref(k) – X(k))$ of claim 24. Rewritten 22.33 is converting the difference of the time domain terms to the frequency domain, instead of subtracting two frequency domain terms. Applicant submits that these differences are a result of the different approaches taught by the DSP handbook and the recitations of claim 24. Withdrawal of the rejection and allowance of claim 24 are respectfully requested.

In the Response to Arguments section of the Office Action dated February 28, 2005, the Examiner states that "the teachings of the DSP Handbook provides a functional relationship of updating the filter weight with the same or similar variables as claimed and finding an optimum relationship of the variables would be an obvious modification for one of ordinary skill in the art of signal processing." (Office Action dated February 28, 2005, page 11). Since the Office Action cites no source for the statements that finding an optimum relationship of the variables would be an obvious modification for one of ordinary skill in the art of signal processing, and

that "similar variables as claimed" could be used for providing a functional relationship of updating the filter weight, such statements must be based upon personal knowledge. Thirty-seven C.F.R. §1.104(d)(2) states that:

> When a rejection in an application is based on facts within the personal knowledge of an employee of the Office, the data shall be as specific as possible, and the reference must be supported, when called for by the applicant, by an affidavit of such employee, and such affidavit shall be subject to contradiction or explanation by the affidavits of the applicant and other persons. (C.F.R. §1.104(d)(2))

If the above rejection is maintained, Representative for Applicant, at this time, requests an affidavit of the Examiner to support the Examiner's statement pursuant to 37 C.F.R. §1.104(d)(2).

Claim 28 recites a system for improving detection of speech signals comprising means for converting reference microphone data from a reference microphone to a frequency domain and means for updating filter weight values with the reference microphone data in the frequency domain. For the reasons described above with regard to claim 21, claim 28 is also patentable over Marash, alone or in combination with the DSP Handbook. Withdrawal of the rejection of claim 28, as well as claims 29-33 which depend therefrom, is respectfully requested.

Claim 31 recites that the updating filter weight value comprises $W(k+1) = W(k)+\mu(Ref(k) - X(k))*conj(Y)$ where k is the data block number and μ is a small adaptive step constant. For the same reasons described above with regard to claim 24, claim 31 is patentable over the Marash, alone or in combination with the DSP Handbook. Withdrawal of the rejection and allowance of claim 31 is respectfully requested.

For the reasons described above, claims 21-33 are patentable over the cited art, and their allowance is respectfully requested.

IV. CONCLUSION

In view of the foregoing remarks, Applicant respectfully submits that the present application is in condition for allowance. Applicant respectfully requests reconsideration of this application and that the application be passed to issue.

Should the Examiner have any questions concerning this paper, the Examiner is invited and encouraged to contact Applicant's undersigned attorney at (216) 621-2234, Ext. 106.

Please charge any deficiency or credit any overpayment in the fees for this amendment to our Deposit Account No. 20-0090.

Respectfully submitted,

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